

05-1-00

A

Practitioner's Docket No. 442-009400-US(PAR)

PATENT

## Preliminary Classification:

Proposed Class:

Subclass:

NOTE: "All applicants are requested to include a preliminary classification on newly filed patent applications. The preliminary classification, preferably class and subclass designations, should be identified in the upper right-hand corner of the letter of transmittal accompanying the application papers, for example 'Proposed Class 2, subclass 129.'" M.P.E.P. § 601, 7th ed.

## IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Box Patent Application  
Assistant Commissioner for Patents  
Washington, D.C. 20231

## NEW APPLICATION TRANSMITTAL

Transmitted herewith for filing is the patent application of

Inventor(s): Miska HANNUKSELA

WARNING: 37 C.F.R. § 1.41(a)(1) points out:

"(a) A patent is applied for in the name or names of the actual inventor or inventors.

"(1) The inventorship of a nonprovisional application is that inventorship set forth in the oath or declaration as prescribed by § 1.63, except as provided for in § 1.53(d)(4) and § 1.63(d). If an oath or declaration as prescribed by § 1.63 is not filed during the pendency of a nonprovisional application, the inventorship is that inventorship set forth in the application papers filed pursuant to § 1.53(b), unless a petition under this paragraph accompanied by the fee set forth in § 1.17(i) is filed supplying or changing the name or names of the inventor or inventors."

For (title):

DATA TRANSMISSION

## CERTIFICATION UNDER 37 C.F.R. § 1.10\*

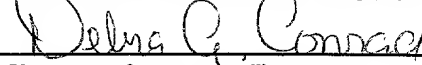
(Express Mail label number is mandatory.)

(Express Mail certification is optional.)

I hereby certify that this New Application Transmittal and the documents referred to as attached therein are being deposited with the United States Postal Service on this date April 28, 2000, in an envelope as "Express Mail Post Office to Addressee," mailing Label Number EL336863045US, addressed to the: Assistant Commissioner for Patents, Washington, D.C. 20231.

Debra G. Conrad

(type or print name of person mailing paper)



Signature of person mailing paper

WARNING: Certificate of mailing (first class) or facsimile transmission procedures of 37 C.F.R. § 1.8 cannot be used to obtain a date of mailing or transmission for this correspondence.

\*WARNING: Each paper or fee filed by "Express Mail" must have the number of the "Express Mail" mailing label placed thereon prior to mailing. 37 C.F.R. § 1.10(b).

"Since the filing of correspondence under § 1.10 without the Express Mail mailing label thereon is an oversight that can be avoided by the exercise of reasonable care, requests for waiver of this requirement will not be granted on petition." Notice of Oct. 24, 1996, 60 Fed. Reg. 56,439, at 56,442.

(New Application Transmittal [4-1]—page 1 of 11)

04/28/00  
JC806 U.S. PTO

JC525 U.S. PTO  
09/560217  
04/28/00

05-1-00

1. Type of Application

This new application is for a(n)

(check one applicable item below)

- ☒ Original (nonprovisional)  
☐ Design  
☐ Plant

**WARNING:** Do not use this transmittal for a completion in the U.S. of an International Application under 35 U.S.C. § 371(c)(4), unless the International Application is being filed as a divisional, continuation or continuation-in-part application.

**WARNING:** Do not use this transmittal for the filing of a provisional application.

**NOTE:** If one of the following 3 items apply, then complete and attach **ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF A PRIOR U.S. APPLICATION CLAIMED** and a **NOTIFICATION IN PARENT APPLICATION OF THE FILING OF THIS CONTINUATION APPLICATION**.

- ☐ Divisional.  
☐ Continuation.  
☐ Continuation-in-part (C-I-P).

2. Benefit of Prior U.S. Application(s) (35 U.S.C. §§ 119(e), 120, or 121)

**NOTE:** A nonprovisional application may claim an invention disclosed in one or more prior filed copending nonprovisional applications or copending international applications designating the United States of America. In order for a nonprovisional application to claim the benefit of a prior filed copending nonprovisional application or copending international application designating the United States of America, each prior application must name as an inventor at least one inventor named in the later filed nonprovisional application and disclose the named inventor's invention claimed in at least one claim of the later filed nonprovisional application in the manner provided by the first paragraph of 35 U.S.C. § 112. Each prior application must also be:

(i) An international application entitled to a filing date in accordance with PCT Article 11 and designating the United States of America; or

(ii) Complete as set forth in § 1.51(b); or

(iii) Entitled to a filing date as set forth in § 1.53(b) or § 1.53(d) and include the basic filing fee set forth in § 1.16; or

(iv) Entitled to a filing date as set forth in § 1.53(b) and have paid therein the processing and retention fee set forth in § 1.21(f) within the time period set forth in § 1.53(f).

37 C.F.R. § 1.78(a)(1).

**NOTE:** If the new application being transmitted is a divisional, continuation or a continuation-in-part of a parent case, or where the parent case is an International Application which designated the U.S., or benefit of a prior provisional application is claimed, then check the following item and complete and attach **ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED**.

**WARNING:** If an application claims the benefit of the filing date of an earlier filed application under 35 U.S.C. §§ 120, 121 or 365(c), the 20-year term of that application will be based upon the filing date of the earliest U.S. application that the application makes reference to under 35 U.S.C. §§ 120, 121 or 365(c). (35 U.S.C. § 154(a)(2) does not take into account, for the determination of the patent term, any application on which priority is claimed under 35 U.S.C. §§ 119, 365(a) or 365(b).) For a c-i-p application, applicant should review whether any claim in the patent that will issue is supported by an earlier application and, if not, the applicant should consider canceling the reference to the earlier filed application. The term of a patent is not based on a claim-by-claim approach. See Notice of April 14, 1995, 60 Fed. Reg. 20,195, at 20,205.

**WARNING:** When the last day of pendency of a provisional application falls on a Saturday, Sunday, or Federal holiday within the District of Columbia, any nonprovisional application claiming benefit of the provisional application must be filed prior to the Saturday, Sunday, or Federal holiday within the District of Columbia. See 37 C.F.R. § 1.78(a)(3).

- ☐ The new application being transmitted claims the benefit of prior U.S. application(s). Enclosed are ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED.

### 3. Papers Enclosed

A. Required for filing date under 37 C.F.R. § 1.53(b) (Regular) or 37 C.F.R. § 1.153 (Design) Application

28 Pages of specification

4 Pages of claims

13 Sheets of drawing

**WARNING:** DO NOT submit original drawings. A high quality copy of the drawings should be supplied when filing a patent application. The drawings that are submitted to the Office must be on strong, white, smooth, and non-shiny paper and meet the standards according to § 1.84. If corrections to the drawings are necessary, they should be made to the original drawing and a high-quality copy of the corrected original drawing then submitted to the Office. Only one copy is required or desired. For comments on proposed then-new 37 C.F.R. § 1.84, see Notice of March 9, 1988 (1990 O.G. 57-62).

**NOTE:** "Identifying indicia, if provided, should include the application number or the title of the invention, inventor's name, docket number (if any), and the name and telephone number of a person to call if the Office is unable to match the drawings to the proper application. This information should be placed on the back of each sheet of drawing a minimum distance of 1.5 cm. (5/8 inch) down from the top of the page . . ." 37 C.F.R. § 1.84(c).

(complete the following, if applicable)

- ☐ The enclosed drawing(s) are photograph(s), and there is also attached a "PETITION TO ACCEPT PHOTOGRAPH(S) AS DRAWING(S)." 37 C.F.R. § 1.84(b).
- ☐ formal
- ☐ informal

### B. Other Papers Enclosed

       Pages of declaration and power of attorney

  1   Pages of abstract

       Other

### 4. Additional papers enclosed

- ☐ Amendment to claims
- ☐ Cancel in this applications claims \_\_\_\_\_ before calculating the filing fee. (At least one original independent claim must be retained for filing purposes.)
- ☐ Add the claims shown on the attached amendment. (Claims added have been numbered consecutively following the highest numbered original claims.)
- ☐ Preliminary Amendment
- ☐ Information Disclosure Statement (37 C.F.R. § 1.98)
- ☐ Form PTO-1449 (PTO/SB/08A and 08B)
- ☐ Citations

- ☐ Declaration of Biological Deposit
- ☐ Submission of "Sequence Listing," computer readable copy and/or amendment pertaining thereto for biotechnology invention containing nucleotide and/or amino acid sequence.
- ☐ Authorization of Attorney(s) to Accept and Follow Instructions from Representative
- ☐ Special Comments
- ☐ Other

**5. Declaration or oath (including power of attorney)**

**NOTE:** A newly executed declaration is not required in a continuation or divisional application provided that the prior nonprovisional application contained a declaration as required, the application being filed is by all or fewer than all the inventors named in the prior application, there is no new matter in the application being filed, and a copy of the executed declaration filed in the prior application (showing the signature or an indication thereon that it was signed) is submitted. The copy must be accompanied by a statement requesting deletion of the names of person(s) who are not inventors of the application being filed. If the declaration in the prior application was filed under § 1.47, then a copy of that declaration must be filed accompanied by a copy of the decision granting § 1.47 status or, if a nonsigning person under § 1.47 has subsequently joined in a prior application, then a copy of the subsequently executed declaration must be filed. See 37 C.F.R. §§ 1.63(d)(1)-(3).

**NOTE:** A declaration filed to complete an application must be executed, identify the specification to which it is directed, identify each inventor by full name including family name and at least one given name, without abbreviation together with any other given name or initial, and the residence, post office address and country or citizenship of each inventor, and state whether the inventor is a sole or joint inventor. 37 C.F.R. § 1.63(a)(1)-(4).

- ☐ Enclosed

Executed by

(check all applicable boxes)

- ☐ inventor(s).
- ☐ legal representative of inventor(s).  
37 C.F.R. §§ 1.42 or 1.43.
- ☐ joint inventor or person showing a proprietary interest on behalf of inventor who refused to sign or cannot be reached.

- ☐ This is the petition required by 37 C.F.R. § 1.47 and the statement required by 37 C.F.R. § 1.47 is also attached. See item 13 below for fee.

- ☒ Not Enclosed.

**NOTE:** Where the filing is a completion in the U.S. of an International Application or where the completion of the U.S. application contains subject matter in addition to the International Application, the application may be treated as a continuation or continuation-in-part, as the case may be, utilizing ADDED PAGE FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION CLAIMED.

- ☒ Application is made by a person authorized under 37 C.F.R. § 1.41(c) on behalf of all the above named inventor(s).

(The declaration or oath, along with the surcharge required by 37 C.F.R. § 1.16(e) can be filed subsequently).

- ☐ Showing that the filing is authorized.  
(not required unless called into question. 37 C.F.R. § 1.41(d))

## 6. Inventorship Statement

**WARNING:** If the named inventors are each not the inventors of all the claims an explanation, including the ownership of the various claims at the time the last claimed invention was made, should be submitted.

The inventorship for all the claims in this application are:

☐ The same.

or

☐ Not the same. An explanation, including the ownership of the various claims at the time the last claimed invention was made,

☐ is submitted.

☐ will be submitted.

## 7. Language

**NOTE:** An application including a signed oath or declaration may be filed in a language other than English. An English translation of the non-English language application and the processing fee of \$130.00 required by 37 C.F.R. § 1.17(k) is required to be filed with the application, or within such time as may be set by the Office. 37 C.F.R. § 1.52(d).

☒ English

☐ Non-English

☐ The attached translation includes a statement that the translation is accurate. 37 C.F.R. § 1.52(d).

## 8. Assignment

☒ An assignment of the invention to Nokia Mobile Phones Ltd.

☐ is attached. A separate ☐ "COVER SHEET FOR ASSIGNMENT (DOCUMENT) ACCOMPANYING NEW PATENT APPLICATION" or ☐ FORM PTO 1595 is also attached.

☒ will follow.

**NOTE:** "If an assignment is submitted with a new application, send two separate letters—one for the application and one for the assignment." Notice of May 4, 1990 (1114 O.G. 77-78).

**WARNING:** A newly executed "CERTIFICATE UNDER 37 C.F.R. § 3.73(b)" must be filed when a continuation-in-part application is filed by an assignee. Notice of April 30, 1993, 1150 O.G. 62-64.

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## 9. Certified Copy

Certified copy(ies) of application(s)

Country	Appln. No.	Filed
Finland	990970	29 April 1999

Country	Appln. No.	Filed
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Country	Appln. No.	Filed
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from which priority is claimed

☐ Is (are) attached.

☒ will follow.

NOTE: The foreign application forming the basis for the claim for priority must be referred to in the oath or declaration. 37 C.F.R. § 1.55(a) and 1.63.

NOTE: This item is for any foreign priority for which the application being filed directly relates. If any parent U.S. application or International Application from which this application claims benefit under 35 U.S.C. § 120 is itself entitled to priority from a prior foreign application, then complete item 18 on the ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED.

## 10. Fee Calculation (37 C.F.R. § 1.16)

A. ☒ Regular application

CLAIMS AS FILED				
Number filed		Number Extra	Rate	Basic Fee 37 C.F.R. § 1.16(a) \$ 690.00
Total				
Claims (37 C.F.R. § 1.16(c))	26 - 20 =	6	× \$ 18.00	108.00
Independent				
Claims (37 C.F.R. § 1.16(b))	6 - 3 =	3	× \$ 78.00	234.00
Multiple dependent claim(s),				
if any (37 C.F.R. § 1.16(d))			+ \$260.00	

☐ Amendment cancelling extra claims is enclosed.

☐ Amendment deleting multiple-dependencies is enclosed.

☐ Fee for extra claims is not being paid at this time.

NOTE: If the fees for extra claims are not paid on filing they must be paid or the claims cancelled by amendment, prior to the expiration of the time period set for response by the Patent and Trademark Office in any notice of fee deficiency. 37 C.F.R. § 1.16(d).

Filing Fee Calculation \$ 1,032.00

B. ☐ Design application  
(\$310.00—37 C.F.R. § 1.16(f))

Filing Fee Calculation \$

C. ☐ Plant application  
(\$480.00—37 C.F.R. § 1.16(g))

Filing fee calculation \$

**11. Small Entity Statement(s)**

- ☐ Statement(s) that this is a filing by a small entity under 37 C.F.R. § 1.9 and 1.27 is (are) attached.

**WARNING:** "Status as a small entity must be specifically established in each application or patent in which the status is available and desired. Status as a small entity in one application or patent does not affect any other application or patent, including applications or patents which are directly or indirectly dependent upon the application or patent in which the status has been established. The refiling of an application under § 1.53 as a continuation, division, or continuation-in-part (including a continued prosecution application under § 1.53(d)), or the filing of a reissue application requires a new determination as to continued entitlement to small entity status for the continuing or reissue application. A nonprovisional application claiming benefit under 35 U.S.C. § 119(e), 120, 121, or 365(c) of a prior application, or a reissue application may rely on a statement filed in the prior application or in the patent if the nonprovisional application or the reissue application includes a reference to the statement in the prior application or in the patent or includes a copy of the statement in the prior application or in the patent and status as a small entity is still proper and desired. The payment of the small entity basic statutory filing fee will be treated as such a reference for purposes of this section." 37 C.F.R. § 1.28(a)(2).

**WARNING:** "Small entity status must not be established when the person or persons signing the . . . statement can unequivocally make the required self-certification." M.P.E.P., § 509.03, 6th ed., rev. 2, July 1996 (emphasis added).

(complete the following, if applicable)

- ☐ Status as a small entity was claimed in prior application  
\_\_\_\_\_ / \_\_\_\_\_, filed on \_\_\_\_\_, from which benefit  
is being claimed for this application under:

35 U.S.C. § ☐ 119(e),  
☐ 120,  
☐ 121,  
☐ 365(c),

and which status as a small entity is still proper and desired.

- ☐ A copy of the statement in the prior application is included.

Filing Fee Calculation (50% of A, B or C above)

\$ \_\_\_\_\_

**NOTE:** Any excess of the full fee paid will be refunded if small entity status is established and a refund request are filed within 2 months of the date of timely payment of a full fee. The two-month period is not extendable under § 1.136. 37 C.F.R. § 1.28(a).

**12. Request for International-Type Search (37 C.F.R. § 1.104(d))**

(complete, if applicable)

- ☐ Please prepare an international-type search report for this application at the time when national examination on the merits takes place.

13. Fee Payment Being Made at This Time

☐ Not Enclosed

☐ No filing fee is to be paid at this time.

*(This and the surcharge required by 37 C.F.R. § 1.16(e) can be paid subsequently.)*

☒ Enclosed

☒ Filing fee \$ 1,032.00

☐ Recording assignment  
(\$40.00; 37 C.F.R. § 1.21(h))  
(See attached "COVER SHEET FOR  
ASSIGNMENT ACCOMPANYING NEW  
APPLICATION".) \$ \_\_\_\_\_

☐ Petition fee for filing by other than all the  
inventors or person on behalf of the inventor  
where inventor refused to sign or cannot be  
reached  
(\$130.00; 37 C.F.R. §§ 1.47 and 1.17(i)) \$ \_\_\_\_\_

☐ For processing an application with a  
specification in  
a non-English language  
(\$130.00; 37 C.F.R. §§ 1.52(d) and 1.17(k)) \$ \_\_\_\_\_

☐ Processing and retention fee  
(\$130.00; 37 C.F.R. §§ 1.53(d) and 1.21(l)) \$ \_\_\_\_\_

☐ Fee for international-type search report  
(\$40.00; 37 C.F.R. § 1.21(e)) \$ \_\_\_\_\_

NOTE: 37 C.F.R. § 1.21(l) establishes a fee for processing and retaining any application that is abandoned for failing to complete the application pursuant to 37 C.F.R. § 1.53(f) and this, as well as the changes to 37 C.F.R. §§ 1.53 and 1.78(a)(1), indicate that in order to obtain the benefit of a prior U.S. application, either the basic filing fee must be paid, or the processing and retention fee of § 1.21(l) must be paid, within 1 year from notification under § 53(f).

Total fees enclosed \$ 1,032.00

14. Method of Payment of Fees

☒ Check in the amount of \$ 1,032.00

☐ Charge Account No. \_\_\_\_\_ in the amount of  
\$ \_\_\_\_\_

A duplicate of this transmittal is attached.

NOTE: Fees should be itemized in such a manner that it is clear for which purpose the fees are paid. 37 C.F.R. § 1.22(b).



## 15. Authorization to Charge Additional Fees

**WARNING:** If no fees are to be paid on filing, the following items should not be completed.

**WARNING:** Accurately count claims, especially multiple dependent claims, to avoid unexpected high charges, if extra claim charges are authorized.

- ☒ The Commissioner is hereby authorized to charge the following additional fees by this paper and during the entire pendency of this application to Account No. 16-1350 :

☒ 37 C.F.R. § 1.16(a), (f) or (g) (filing fees)

☒ 37 C.F.R. § 1.16(b), (c) and (d) (presentation of extra claims)

**NOTE:** Because additional fees for excess or multiple dependent claims not paid on filing or on later presentation must only be paid or these claims cancelled by amendment prior to the expiration of the time period set for response by the PTO in any notice of fee deficiency (37 C.F.R. § 1.16(d)), it might be best not to authorize the PTO to charge additional claim fees, except possibly when dealing with amendments after final action.

☒ 37 C.F.R. § 1.16(e) (surcharge for filing the basic filing fee and/or declaration on a date later than the filing date of the application)

☒ 37 C.F.R. § 1.17(a)(1)-(5) (extension fees pursuant to § 1.136(a)).

☐ 37 C.F.R. § 1.17 (application processing fees)

**NOTE:** ". . . A written request may be submitted in an application that is an authorization to treat any concurrent or future reply, requiring a petition for an extension of time under this paragraph for its timely submission, as incorporating a petition for extension of time for the appropriate length of time. An authorization to charge all required fees, fees under § 1.17, or all required extension of time fees will be treated as a constructive petition for an extension of time in any concurrent or future reply requiring a petition for an extension of time under this paragraph for its timely submission. Submission of the fee set forth in § 1.17(a) will also be treated as a constructive petition for an extension of time in any concurrent reply requiring a petition for an extension of time under this paragraph for its timely submission." 37 C.F.R. § 1.136(a)(3).

☐ 37 C.F.R. § 1.18 (issue fee at or before mailing of Notice of Allowance, pursuant to 37 C.F.R. § 1.311(b))

**NOTE:** Where an authorization to charge the issue fee to a deposit account has been filed before the mailing of a Notice of Allowance, the issue fee will be automatically charged to the deposit account at the time of mailing the notice of allowance. 37 C.F.R. § 1.311(b).

**NOTE:** 37 C.F.R. § 1.28(b) requires "Notification of any change in status resulting in loss of entitlement to small entity status must be filed in the application . . . prior to paying, or at the time of paying, . . . the issue fee. . . ." From the wording of 37 C.F.R. § 1.28(b), (a) notification of change of status must be made even if the fee is paid as "other than a small entity" and (b) no notification is required if the change is to another small entity.

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16. Instructions as to Overpayment

NOTE: "... Amounts of twenty-five dollars or less will not be returned unless specifically requested within a reasonable time, nor will the payer be notified of such amounts; amounts over twenty-five dollars may be returned by check or, if requested, by credit to a deposit account." 37 C.F.R. § 1.26(a).

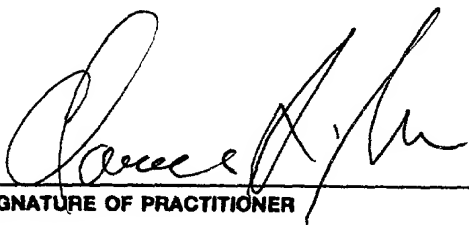
- ☒ Credit Account No. 16-1350  
☐ Refund

SEND ALL CORRESPONDENCE TO:

Reg. No. 24,622

Tel. No. (203) 259-1800

Customer No.

  
SIGNATURE OF PRACTITIONER

Clarence A. Green

(type or print name of attorney)

PERMAN & GREEN, LLP

P.O. Address

425 Post Road, Fairfield, Connecticut 06430

☐ **Incorporation by reference of added pages**

*(check the following item if the application in this transmittal claims the benefit of prior U.S. application(s) (including an international application entering the U.S. stage as a continuation, divisional or C-I-P application) and complete and attach the ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED)*

- ☐ Plus Added Pages for New Application Transmittal Where Benefit of Prior U.S. Application(s) Claimed

Number of pages added \_\_\_\_\_

- ☐ Plus Added Pages for Papers Referred to in Item 4 Above

Number of pages added \_\_\_\_\_

- ☐ Plus added pages deleting names of inventor(s) named in prior application(s) who is/are no longer inventor(s) of the subject matter claimed in this application.

Number of pages added \_\_\_\_\_

- ☐ Plus "Assignment Cover Letter Accompanying New Application"

Number of pages added \_\_\_\_\_

☒ **Statement Where No Further Pages Added**

*(if no further pages form a part of this Transmittal, then end this Transmittal with this page and check the following item)*

- ☒ This transmittal ends with this page.

## DATA TRANSMISSION

### FIELD OF THE INVENTION

- 5 This invention relates to data transmission and is particularly, but not exclusively, related to transmission of data representative of picture sequences, such as video. It is particularly suited to transmission over links susceptible to errors and loss of data.

### 10 BACKGROUND OF THE INVENTION

- During the past few years, the amount of multimedia content available through the Internet has increased considerably. Since data delivery rates to mobile terminals are becoming high enough to enable such terminals to be able to retrieve multimedia content, it is becoming desirable to enable mobile terminals to retrieve video and other multimedia content from the Internet. An example of a high-speed data delivery system is the upcoming GSM phase 2+.
- 15

- The term multimedia as used herein applies to both sound and pictures, to sound only and to pictures only. Sound may include speech and music.
- 20

- Network traffic through the Internet is based on a transport protocol called the Internet Protocol (IP). IP is concerned with transporting data packets from one location to another. It facilitates the routing of packets through intermediate gateways, that is, it allows data to be sent to machines that are not directly connected in the same physical network. The unit of data transported by the IP layer is called an IP datagram. The delivery service offered by IP is connectionless, that is IP datagrams are routed around the Internet independently of each other. Since no resources are permanently committed within the gateways to any particular connection, the gateways may occasionally have to discard datagrams because of lack of buffer space or other resources. Thus, the delivery service offered by IP is a best effort service rather than a guaranteed service.
- 25
- 30

Internet multimedia is typically streamed over the User Datagram Protocol (UDP), the Transmission Control Protocol (TCP) or the Hypertext Transfer Protocol (HTTP).

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UDP is a connectionless lightweight transport protocol. It offers very little above the service offered by IP. Its most important function is to deliver datagrams between specific transport endpoints. Consequently, the transmitting application has to take care of how to packetize data to datagrams. Headers used in UDP contain a checksum that allows the UDP layer at the receiving end to check the validity of the data. Otherwise, degradation of IP datagrams will in turn affect UDP datagrams. UDP does not check that the datagrams have been received, does not retransmit missing datagrams, nor does it guarantee that the datagrams are received in the same order as they were transmitted.

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UDP introduces a relatively stable throughput having a small delay since there are no retransmissions. Therefore it is used in retrieval applications to deal with the effect of network congestion and to reduce delay (and jitter) at the receiving end. However, the client must be able to recover from packet losses and possibly conceal lost content. Even with reconstruction and concealment, the quality of a reconstructed clip suffers somewhat. On the other hand, playback of the clip is likely to happen in real-time without annoying pauses. Firewalls, whether in a company or elsewhere, may forbid the usage of UDP because it is connectionless.

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TCP is a connection-orientated transport protocol and the application using it can transmit or receive a series of bytes with no apparent boundaries as in UDP. The TCP layer divides the byte stream into packets, sends the packets over an IP network and ensures that the packets are error-free and received in their correct order. The basic idea of how TCP works is as follows. Each time TCP sends a packet of data, it starts a timer. When the receiving end gets the packet, it immediately sends an acknowledgement back to the sender. When the sender receives the acknowledgement, it knows all is well, and cancels the timer.

30

However, if the IP layer loses the outgoing segment or the return acknowledgement, the timer at the sending end will expire. At this point, the sender will retransmit the segment. Now, if the sender waited for an acknowledgement for each packet before sending the next one, the overall transmission time would be relatively long and dependent on the round-trip delay between the sender and the receiver. To overcome this problem, TCP uses a sliding window protocol that allows several unacknowledged packets to be present in the network. In this protocol, an acknowledgement packet contains a field filled with the number of bytes the client is willing to accept (beyond the ones that are currently acknowledged). This window size field indicates the amount of buffer space available at the client for storage of incoming data. The sender may transmit data within the limit indicated by the latest received window size field. The sliding window protocol means that TCP effectively has a slow start mechanism. At the beginning of a connection, the very first packet has to be acknowledged before the sender can send the next one. Typically, the client then increases the window size exponentially. However, if there is congestion in the network, the window size is decreased (in order to avoid congestion and to avoid receive buffer overflow). The details how the window size is changed depend on the particular TCP implementation in use.

A multimedia content creation and retrieval system is shown in Figure 1. The system has one or more media sources, for example a camera and a microphone. Alternatively, multimedia content can also be synthetically created without a natural media source, for example animated computer graphics and digitally generated music. In order to compose a multimedia clip consisting of different media types, such as video, audio, text, images, graphics and animation, raw data captured from the sources are edited by an editor. Typically the storage space taken up by raw (uncompressed) multimedia data is huge. It can be megabytes for a video sequence which can include a mixture of different media, for example animation. In order to provide an attractive multimedia retrieval service over low bit rate channels, for example 28.8 kbps and 56 kbps, multimedia clips are compressed in the editing phase. This typically occurs off-line. The clips are then

handed to a multimedia server. Typically, a number of clients can access the server over one or more networks. The server is able to respond to the requests presented by the clients. The main task of the server is to transmit a desired multimedia clip to the client which the client decompresses and plays. During playback, the client utilizes one or more output devices, such as a screen and a loudspeaker. In some circumstances, clients are able to start playback while data are still being downloaded.

It is convenient to deliver a clip by using a single channel which provides a similar quality of service for the entire clip. Alternatively different channels can be used to deliver different parts of a clip, for example sound on one channel and pictures on another. Different channels may provide different qualities of service. In this context, quality of service includes bit rate, loss or bit error rate and transmission delay variation.

In order to ensure multimedia content of a sufficient quality is delivered, it is provided over a reliable network connection, such as TCP, which ensures that received data are error-free and in the correct order. Lost or corrupted protocol data units are retransmitted. Consequently, the channel throughput can vary significantly. This can even cause pauses in the playback of a multimedia stream whilst lost or corrupted data are retransmitted. Pauses in multimedia playback are annoying.

Sometimes retransmission of lost data is not handled by the transport protocol but rather by some higher-level protocol. Such a protocol can select the most vital lost parts of a multimedia stream and request the retransmission of those. The most vital parts can be used for prediction of other parts of the stream, for example.

In order to understand the invention better, descriptions of the elements of the retrieval system, namely the editor, the server and the client, are set out below.

A typical sequence of operations carried out by the multimedia clip editor is shown in Figure 2. Raw data are captured from one or more data sources. Capturing is done using hardware, device drivers dedicated to the hardware and a capturing application which controls the device drivers to use the hardware. Capturing hardware may consist of a video camera connected to a PC video grabber card, for example. The output of the capturing phase is usually either uncompressed data or slightly compressed data with irrelevant quality degradations when compared to uncompressed data. For example, the output of a video grabber card could be in an uncompressed YUV 4:2:0 format or in a motion-JPEG format.

The YUV colour model and the possible sub-sampling schemes are defined in Recommendation ITU-R BT.601-5 "Studio Encoding Parameters of Digital Television for Standard 4:3 and Wide-Screen 16:9 Aspect Ratios". Relevant digital picture formats such as CIF, QCIF and SQCIF are defined in Recommendation ITU-T H.261 "Video Codec for Audiovisual Services at p x 64 kbits" (section 3.1 "Source Formats").

During editing separate media tracks are tied together in a single timeline. It is also possible to edit the media tracks in various ways, for example to reduce the video frame rate. Each media track may be compressed. For example, the uncompressed YUV 4:2:0 video track could be compressed using ITU-T recommendation H.263 for low bit rate video coding. If the compressed media tracks are multiplexed, they are interleaved so that they form a single bitstream. This clip is then handed to the multimedia server. Multiplexing is not essential to provide a bitstream. For example, different media components such as sounds and images may be identified with packet header information in the transport layer. Different UDP port numbers can be used for different media components.

A typical sequence of operations carried out by the multimedia server is shown in Figure 3. Typically multimedia servers have two modes of operation; they deliver either pre-stored multimedia clips or a live (real-time) multimedia stream. In the first mode, clips are stored in a server database, which is then accessed on-demand by the server. In the second mode, multimedia clips are handed to the



server as a continuous media stream that is immediately transmitted to clients. Clients control the operation of the server by an appropriate control protocol being at least able to select a desired media clip. In addition, servers may support more advanced controls. For example, clients may be able to stop the transmission of a clip, to pause and resume transmission of a clip, and to control the media flow in case of a varying throughput of the transmission channel in which case the server must dynamically adjust the bitstream to fit into the available bandwidth.

A typical sequence of operations carried out by the multimedia retrieval client is shown in Figure 4. The client gets a compressed and multiplexed media clip from a multimedia server. The client demultiplexes the clip in order to obtain separate media tracks. These media tracks are then decompressed to provide reconstructed media tracks which are played out with output devices. In addition to these operations, a controller unit is provided to interface with end-users, that is to control playback according to end-user input and to handle client-server control traffic. It should be noted that the demultiplexing-decompression-playback chain can be done on a first part of the clip while still downloading a subsequent part of the clip. This is commonly referred to as streaming. An alternative to streaming is to download the whole clip to the client and then demultiplex it, decompress it and play it.

A typical approach to the problem of varying throughput of a channel is to buffer media data in the client before starting the playback and/or to adjust the transmitted bit rate in real-time according to channel throughput statistics.

One way of tackling the problem of pauses is by using dynamic bit rate adjustment in the multimedia server. However, the server may not react to network congestion sufficiently quickly to avoid pausing in the client. In addition, the server cannot control the retransmission mechanism of TCP (or other underlying protocols, such as IP).

Even if dynamic bit rate adjustment is used, the client has to do some initial buffering in any case to avoid delivery delays caused by re-transmission. If a constant channel bit rate is assumed, one can calculate the point in time at which a data unit is supposed to have been completely received. In addition, one can

5 calculate the point in time by which a data unit is supposed to have been played. The time difference between these two points in time is referred to as the safety time. Another way of defining the safety time is to state that it is the maximum time between two consecutively received data units which does not cause pausing in the playback.

10 When calculating the safety times for a clip, each data unit has to be considered separately. The calculations assume that no throughput drops occur before the data unit that is currently being processed. If the maximum throughput of the channel is equal to the average bit rate of the multimedia clip, the client cannot

15 recover from a drop in the amount of received bits after throughput has dropped. The only way to guarantee some protection against throughput drops is to buffer some data before starting playback. If the channel stops providing data, the client can still play the stream while there are data in a buffer. Thus, the average safety time is approximately equal to the initial buffering time. Since the bit rate of the clip

20 may vary, the safety time also may vary and the minimum safety time is thus equal to or less than the initial buffering delay.

HTTP, the Hypertext Transfer Protocol, is the basis of the World Wide Web (WWW). It is a simple protocol. A client establishes a TCP connection to a server,

25 issues a request, and reads back the server's response. The server denotes the end of its response by closing the connection. The arrangement of protocol layers is typically HTTP on TCP which is on IP.

The most common HTTP request is called GET. The GET request is associated

30 with a universal resource identifier (URI) which uniquely specifies the requested item. The server responds to the GET request by returning the file corresponding to the specified URI. The file returned by the server normally contains pointers

(hypertext links) to other files that can reside on other servers. A user can therefore easily follow the links from file to file.

Servers used for Internet multimedia retrieval are either dedicated multimedia  
5 servers or normal WWW servers.

Dedicated multimedia servers are typically capable of transmission over HTTP, TCP and UDP protocols. They may be able to readjust the contents of media clips dynamically to meet the available network bandwidth and to avoid network  
10 congestion. They may also support fast forward and fast rewind operations as well as live multimedia streaming. They can provide a number of streams simultaneously.

Multimedia servers based on normal WWW servers are also referred to as server-  
15 less or HTTP multimedia solutions. The multimedia clips are streamed over HTTP. Since this type of server has no control over the contents of the stream, no flow (bandwidth) control can be applied, and it cannot respond to network congestion. Therefore, sudden pauses in the playback can occur. Consequently there must be a relatively long initial buffering delay in the client before starting the playback to  
20 avoid such sudden pauses. Fast-forwarding a multimedia stream from a standard WWW server is not possible. Live multimedia streaming must be implemented using special tricks, such as Java programming.

When a streamed multimedia clip is received, a suitable independent media player  
25 application or a browser plug-in can be used to play it. Such multimedia players differ largely from browser to browser. Newer browsers may have some integrated plug-ins for the most popular streaming video players.

There are a number of different data transmission methods available for  
30 transmitting data between mobile terminals and their networks. One of the best known methods is GSM (Global System for Mobile communications).

The current GSM data service called Circuit Switched Data (CSD) offers a 9.6 kbps circuit-switched channel. It is intended for GSM to provide a 14.4 kbps data channel having forward error correction (FEC) and status information. High Speed Circuit Switched Data (HSCSD) provides multiple 9.6 kbps or 14.4 kbps time slots for a single user at the same time. There are symmetric and asymmetric connections. In a symmetric connection, the air interface resources are allocated symmetrically, and provide the same data transmission rate in both directions. In an asymmetric connection, different data rates are supported for up-link and for down-link. However, asymmetric air interface connections are applicable only in non-transparent mode (see below).

Circuit-switched GSM data systems, CSD and HSCSD, offer two basic connection types, namely transparent (T) and non-transparent (NT). These are distinguished by the way they correct transmission errors. In a transparent connection, error correction is done solely by a forward error correction mechanism provided by the radio interface transmission scheme. The connection is considered as a synchronous circuit. The available throughput is constant, and the transmission delay is fixed. The transmitted data are likely to contain bit inversion errors. In a non-transparent connection, the GSM circuit connection is regarded as a packet (or frame) data flow although the end-to-end service is circuit-switched. Each frame includes redundancy bits to enable a receiver to detect remaining errors. There are two sources of error, packet drops and corruption due to interference in the radio frequency path. The latter can be recovered by redundancy checking. The Radio Link Protocol (RLP) is used to provide retransmission in case of remaining errors in frame. If a frame is found to be correct, the receiver acknowledges this fact. If it is found not to be correct, a negative acknowledgement is sent and the indicated frame is retransmitted. Consequently, a non-transparent connection is error-free but throughput and transmission delay vary.

Other network types exist, for example GPRS (General Packet Radio System). In GPRS the transmissions are truly packet based.

A video sequence consists of a series of still pictures. Video compression methods are based on reducing redundant and perceptually irrelevant parts of video sequences. The redundancy in video sequences can be categorised into spatial, temporal and spectral redundancy. Spatial redundancy means the correlation between neighbouring pixels. Temporal redundancy means that the same objects appear in consecutive pictures. Reducing the temporal redundancy reduces the amount of data required to represent a particular image sequence and thus compresses the data. This can be achieved by generating motion compensation data, which describe the motion between the current and a previous (reference) picture. In effect, the current picture is predicted from the previous one. Spectral redundancy means the correlation between the different colour components of the same picture.

Simply reducing the redundancy of the sequence does not usually compress it enough. Therefore, some video encoders try to reduce the quality of those parts of a video sequence which are subjectively the least important. In addition, the redundancy of the encoded bitstream is reduced by means of efficient lossless coding of compression parameters and coefficients. The main technique is to use variable length codes.

Video compression methods typically differentiate between pictures that can use temporal redundancy reduction and those that cannot. Compressed pictures, which do not use temporal redundancy reduction methods, are usually called INTRA or I-frames whereas temporally predicted pictures are called INTER or P-frames. In the INTER frame case, the predicted (motion-compensated) picture is rarely precise enough, and therefore a spatially compressed prediction error picture is also associated with each INTER frame.

Temporal scalability provides a mechanism for enhancing perceptual quality by increasing the picture display rate. This is achieved by taking a pair of consecutive reference pictures and bi-directionally predicting a B-picture from either one or

both of them. The B-picture can then be played in sequence between the two anchor pictures. This is illustrated in Figure 5. Bi-directional temporal prediction yields a more accurately predicted picture than uni-directional prediction. Thus, for the same quantization level, B-pictures yield increased compression as compared to forwardly predicted P-pictures. B-pictures are not used as reference pictures, that is other pictures are never predicted from them. Since they can be discarded without impacting the picture quality of future pictures, they provide temporal scalability. It should be noted that while B-pictures provide better compression performance than P-pictures, they are more complex to construct and require more memory. Furthermore they introduce additional delays because bi-directional interpolation requires both reference pictures to have been received and additional calculations are required. In addition, B-pictures require more side information in the bitstream.

The term scalability refers to the capability of a compressed sequence to be decoded at different data rates. In other words, a scalable multimedia clip can be edited relatively easily while it is compressed so that it can be streamed over channels with different bandwidths and still be decoded and played back in real-time.

Scalable multimedia is typically ordered so that there are hierarchical layers of data. A base layer contains a basic representation of the multimedia clip whereas enhancement layers contain refinement data on top of underlying layers. Consequently, the enhancement layers improve the quality of the clip.

Scalability is a desirable property for heterogeneous and error prone environments. This property is desirable in order to counter limitations such as constraints on bit rate, display resolution, network throughput, and decoder complexity.

Scalability can be used to improve error resilience in a transport system where layered coding is combined with transport prioritisation. The term transport

prioritisation here refers to various mechanisms to provide different qualities of service in transport, including unequal error protection, to provide different channels having different error/loss rates. Depending on their nature, data are assigned differently, for example, the base layer may be delivered through a  
 5 channel with high degree of error protection, and the enhancement layers may be transmitted through more error-prone channels.

Generally, scalable multimedia coding suffers from a worse compression efficiency than non-scalable coding. In other words, a multimedia clip coded as a scalable  
 10 multimedia clip with all enhancement layers requires greater bandwidth than if it had been coded as a non-scalable single-layer clip with equal quality. However, exceptions to this general rule exist, for example the temporally scalable B-frames in video compression.

15 In the following, scalability is discussed with reference to the ITU-T H.263 video compression standard. H.263 is an ITU-T recommendation for video coding in low bit rate communication which generally means data rates below 64 kbps. The recommendation specifies the bitstream syntax and the decoding of the bitstream. Currently, there are two versions of H.263. Version 1 consists of the core algorithm  
 20 and four optional coding modes. H.263 version 2 is an extension of version 1 providing twelve new negotiable coding modes.

Pictures are coded as luminance and two colour difference (chrominance) components ( $Y$ ,  $C_B$  and  $C_R$ ). The chrominance pictures are sampled with a half of  
 25 the pixels along the both co-ordinate axes when compared to the luminance picture.

The scalability mode (Annex O) of H.263 specifies syntax to support temporal, signal-to noise ratio (SNR), and spatial scalability capabilities.

30 Spatial scalability and SNR scalability are closely related, the only difference being the increased spatial resolution provided by spatial scalability. An example of SNR

scalable pictures is shown in Figure 6. SNR scalability implies the creation of multi-rate bit streams. It allows for the recovery of coding errors, or differences between an original picture and its reconstruction. This is achieved by using a finer quantizer to encode the difference picture in an enhancement layer. This additional information increases the SNR of the overall reproduced picture.

Spatial scalability allows for the creation of multi-resolution bit streams to meet varying display requirements and/or constraints. A spatially scalable structure is illustrated in Figure 7. It is essentially the same as in SNR scalability except that a spatial enhancement layer attempts to recover the coding loss between an up-sampled version of the reconstructed reference layer picture and a higher resolution version of the original picture. For example, if the reference layer has a quarter common intermediate format (QCIF) resolution, and the enhancement layer has a common intermediate format (CIF) resolution, the reference layer picture must be scaled accordingly such that the enhancement layer picture can be predicted from it. The QCIF standard allows the resolution to be increased by a factor of two in the vertical direction only, horizontal direction only, or both the vertical and horizontal directions for a single enhancement layer. There can be multiple enhancement layers, each increasing the picture resolution over that of the previous layer. The interpolation filters used to up-sample the reference layer picture are explicitly defined in the H.263 standard. Aside from the up-sampling process from the reference to the enhancement layer, the processing and syntax of a spatially scaled picture are identical to those of an SNR scaled picture.

In either SNR or spatial scalability, the enhancement layer pictures are referred to as EI- or EP-pictures. If the enhancement layer picture is upwardly predicted from a picture in the reference layer, then the enhancement layer picture is referred to as an Enhancement-I (EI) picture. In this type of scalability, the reference layer means the layer "below" the current enhancement layer. In some cases, when reference layer pictures are poorly predicted, over-coding of static parts of the picture can occur in the enhancement layer, causing an unnecessarily excessive bit rate. To avoid this problem, forward prediction is permitted in the enhancement



layer. A picture that can be predicted in the forward direction from a previous enhancement layer picture or, alternatively, upwardly predicted from the reference layer picture is referred to as an Enhancement-P (EP) picture. Note that computing the average of the upwardly and forwardly predicted pictures can provide bi-directional prediction for EP-pictures. For both EI- and EP-pictures, upward prediction from the reference layer picture implies that no motion vectors are required. In the case of forward prediction for EP-pictures, motion vectors are required.

10 In multi-point and broadcast multimedia applications, constraints on network throughput may not be foreseen at the time of encoding. Thus, a scalable bitstream should be used. Figure 8 shows an IP multicasting arrangement where each router can strip the bitstream according to its capabilities. It shows a server S providing a bitstream to a number of clients C. The bitstreams are routed to the clients by routers R. In this example, the server is providing a clip which can be scaled to at least three bit rates, 120 kbit/s, 60 kbit/s and 28 kbit/s.

15 If the client and server are connected via a normal uni-cast connection, the server may try to adjust the bit rate of the transmitted multimedia clip according to the temporary channel throughput. One solution is to use a layered bit stream and to adapt to bandwidth changes by varying the number of transmitted enhancement layers.

## 25 SUMMARY OF THE INVENTION

According to a first aspect of the invention there is provided a method of transmitting a data signal having a sequence of data units in a predetermined order over a transmission link between a data source and a data sink comprising the step of sending the data units in an order determined by their relative importance rather than their predetermined order.

According to a second aspect of the invention there is provided a data transmission system for transmitting a data signal having a sequence of data units in a predetermined order over a transmission link between a data source and a data sink the data units being sent in an order determined by their relative importance rather than their predetermined order.

According to a third aspect of the invention there is provided a server for transmitting a data signal having a sequence of data units in a predetermined order over a transmission link the data units being sent in an order determined by their relative importance rather than their predetermined order.

Preferably the server comprises re-ordering means. Preferably the server comprises an editor for providing a scalable data signal.

According to a fourth aspect of the invention there is provided a computer program product stored on a computer usable medium comprising computer readable program means for causing transmission of a data signal having a sequence of data units in a predetermined order over a transmission link the data units being sent in an order determined by their relative importance rather than their predetermined order.

Preferably the computer program product comprises a server. In this sense it is meant that when the computer program is loaded into hardware it causes the hardware to operate as a server device. Preferably the computer program product comprises an editor for providing a scalable data signal. Preferably the computer program product comprises re-ordering means for re-ordering the data units.

According to a fifth aspect of the invention there is provided a data signal having a sequence of data units for transmission over a transmission link between a data source and a data sink the data units being in an order determined by their relative importance rather than their predetermined order.

Preferably where the data units or layers of the data units are ordered into a sequence other than their original sequence, the invention also includes means or a step for returning the data units or layers of the data units into their original sequence.

5

The transmission link may simply be a bottleneck where congestion is occurring and the invention is used to overcome the bottleneck.

10

An advantage of the invention is that it can provide larger safety times than prior art arrangements with the same initial buffering. In addition, it provides some degree of protection against decreased transmission throughput. Therefore it helps in reducing the occurrence of pauses without sacrificing compression efficiency.

15

The invention also helps in transmitting over a unreliable network connection using transmission protocols which do not automatically retransmit but only retransmit those missing parts chosen by a higher level function, for example in response to feedback. In this case, it provides more time for retransmission of the vital parts of a data signal than prior art arrangements. Thus, vital parts of a stream are more likely to be transmitted successfully and the ultimate end user of the data signal is likely to perceive a better output quality.

20

Preferably the bandwidth of the transmission link available to the data signal is variable. This may be because the amount of data traffic using the link is variable, for example due to the need to re-transmit lost or corrupted data.

25

The invention provides a way of altering a data signal so that it can be transmitted over a transmission link having a bandwidth which would otherwise not be sufficient for it or for allowing certain of the bandwidth of its transmission link to be made available for other data, for example control information.

30

Preferably the effect of the invention is to increase the safety times of some data units relative to, or even at the expense of, the safety times of other data units.

Preferably the data signal is arranged in a hierarchy having a base layer and at least one enhancement layer. The term base layer means a most fundamental layer. The base layer may represent the least refined version of the data signal. In each unit, the (or each) enhancement layer, when combined with the base layer, refines the base layer to make it more like the data unit. Each enhancement layer may add more refinement to the base layer. Preferably the safety time of the base layer is increased relative to the safety times of the enhancement layers.

Preferably the data signal is scalable. The base layer and the or each enhancement layer may have a scaled relationship with respect to each other. The signal may be scalable in the temporal, the spatial, the spectral or the SNR domains.

Preferably by virtue of re-ordering the sequence the layers have safety times which depend on their position in the hierarchy. For example the base layer of a particular data unit may have a greater safety time than enhancement layers. Advantageously, the finer the level of refinement represented by a layer, the shorter its safety time.

Preferably the source is a server. Alternatively it is an editor.

Preferably the sink is a client. It may be a destination of the data signal such as a mobile terminal, for example a mobile telephone.

Preferably the data units are frames. In one embodiment the data units represent a sequence of pictures. Conveniently it is a video sequence. Most preferably, it comprises multimedia data.

It should be understood that the terms "layer" and "layers" are being used figuratively and refer to blocks of data which can be applied to each other rather than to a strictly positional relationship.

## 5 BRIEF DESCRIPTION OF THE DRAWINGS

The invention will now be described by way of example only with reference to the accompanying drawings in which:

Figure 1 shows a multimedia content creation and retrieval system;

10 Figure 2 shows operations of a multimedia clip editor;

Figure 3 shows operations of a multimedia server;

Figure 4 shows operations of a multimedia retrieval client;

Figure 5 shows an illustration of B picture prediction;

Figure 6 shows an illustration of SNR scalability;

15 Figure 7 shows an illustration of spatial scalability;

Figure 8 shows scalable multimedia being used in an IP multicast system;

Figure 9 shows mobile multimedia retrieval system according to the invention;

Figure 10 shows operations of an editor according to the invention;

Figure 11 shows operations of a client according to the invention;

20 Figure 12 shows safety times of a re-ordered sequence and an original sequence;

Figure 13 shows receiving and playback of a re-ordered sequence;

Figure 14 also shows receiving and playback of a re-ordered sequence; and

Figure 15 shows another multimedia content creation and retrieval system.

## 25 DETAILED DESCRIPTION

Figures 1 to 8 have been discussed above.

A mobile multimedia retrieval system 10 according to the invention is shown in

30 Figure 9. Although only a single client 22 is shown it will be appreciated that there can be, and in preferred embodiments will be, a number of such clients. It will also

be appreciated that at least part of the transmission link connecting a server 18 and the client 22 will be wireless.

The system 10 comprises an editor 12 which produces a compressed clip, a server database 14 for storing the compressed clip, re-ordering means 16 for re-ordering the sequence of the compressed clip, a server 18 for sending the re-ordered compressed clip and a client comprising order recovery means 20 to recover the original sequence of the compressed clip. The client 22 controls the operation of the server, for example by specifying which data are sent and how they are sent. Clearly, the re-ordering means 16 and the order recovery means 20 need to be compatible so that the latter can recover the original order. The re-ordering means 16 could be located to operate on data before they reach the server database 14 rather than the embodiment shown in Figure 9. Certain types of data, such as a live stream, may by-pass the database 14 and be sent by the editor 12 directly to the re-ordering means 16. Although they are shown separately in Figure 9, the editor 12, the server database 14, the re-ordering means 16 and the server 18 can all be configured into a single unit or entity which can also be referred to as a server.

Figure 10 shows operation of an editor according to the invention. Raw data captured in the form of one or more uncompressed streams are edited to produce one or more uncompressed tracks. These tracks are compressed in a compression step to produce compressed tracks. The compressed tracks are multiplexed together to produce a compressed multimedia clip. The multimedia clips are coded scalably. The resulting bitstream consists of basic units, such as frames in video compression. These basic units are not all of equal importance; some are primary basic units which are vital to decode the bitstream and some are secondary basic units which improve the quality of the bitstream. Elimination of the secondary basic units causes only a quality loss in the decoded multimedia clip. In effect, the primary basic units are a base layer of the bitstream and the secondary units are enhancement layers.

Instead of organising the basic units into the original transmission order, that is the sequence in which they are produced by editing and compression of raw data, the invented method re-orders their sequence by prioritising the primary basic units so that they pre-empt the secondary basic units in the transmission order. Reordering can take place before or after the multiplexing process depending on the nature of the system and the data. In Figure 10 it is shown as taking place after multiplexing.

A re-ordering algorithm is used to re-order the sequence. In a preferred embodiment, the algorithm conducts stream analysis to identify primary and secondary basic units and to collect bit rate statistics. These may include the number of bits per frame. The bitstream is then re-ordered with iterative re-ordering steps to minimise the initial buffering time needed for full-quality playback with full channel throughput and reduced-quality playback with a reduced channel throughput. An example of such a re-ordering algorithm is given below. It should be noted that the expected reduced channel throughput has to be estimated in advance since it is one of the input parameters of the algorithm. The estimation can be based on the statistics obtained from the channel behaviour, for example.

Re-ordering of the basic units means that the primary basic units are likely to be received earlier than indicated in their playback time even though some decrease in the channel throughput occurs. However, secondary basic units are vulnerable to transmission delay variations since the method schedules them to be received just before their indicated playback time. In this way the client is still likely to be able to play the primary basic units without pausing the playback, even if the channel throughput decreases considerably. The secondary basic units, on the other hand, may arrive later than their scheduled playback time and will not be used in playback.

Figure 11 shows operation of a client according to the invention. The bitstream is received by the client, re-ordered to produce the original compressed clip, demultiplexed to produce the compressed tracks and then decompressed to reconstruct the tracks. The basic unit order recovery step in the client can be

applied either before or after demultiplexing (depending in when it occurs in the content editor). In Figure 11 it is shown as occurring before demultiplexing, to correspond to the order described in Figure 10. In this way re-ordering and order recovery are both applied to the multiplexed bitstream. Order recovery organises the basic units back into their natural sequence for decompression and discards all late-arriving secondary basic units.

On receipt of a basic unit by the client, it is time-stamped with a time by which it should be played. This time-stamp is compared with the current playback time. If the time-stamp is greater than the current playback time the basic unit is not decompressed and it is discarded. If the time-stamp is less than the current playback time the basic unit is decompressed and the two times are compared after decompression. If the time stamp is less than the current playback time the basic unit is played. Otherwise, it is discarded.

In the particular case in which re-ordering is performed after multiplexing, the multiplexing/demultiplexing format used needs to allow re-ordering of the basic units and recovery of the original order.

Figure 15 shows another view of a multimedia content creation and retrieval system according to the invention. The system 50 comprises an editor 52, a server 54 and a client 56.

The editor comprises a compressor 58 for compressing a raw data stream typically being in the form of a sequence of images into a form suitable for transmission, a multiplexer 60 for multiplexing the compressed raw data stream with other compressed data streams together to produce a compressed bitstream and layer generation means 62 for producing primary basic frames which are vital to decode the bitstream and secondary basic frames which improve the quality of the bitstream.



The server 54 comprises re-ordering means 64 for arranging the frames in an order which depends on their importance in helping to reconstruct the raw data stream at the client 56 rather than their natural sequential order.

- 5 The server 54 and the client 56 each comprise transmitting/receiving means 66 and 70 respectively for data transmission over a transmission link 68 between the server 54 and the client 56.

- 10 The client 56 comprises order recovery means 72 for putting the frames back into their natural sequential order, a demultiplexer 74 for demultiplexing the bitstream and a decompressor 76 for decompressing the compressed data streams to reproduce the raw data stream. The client comprises playing/output means 78 for playing or otherwise outputting the raw data stream. The client also comprises a controller 80 which applies a control signal to the server 54 to control the re-ordering means in response to receipt of frames in the client 56.

#### Example

- 20 An embodiment of the invention was tested using an H.263 compliant encoder and a frame re-ordering tool. The encoder was capable of producing B-frames. The re-ordering tool comprised a re-ordering algorithm and could be adjusted to produce frame ordering for different decreased channel data rates.

- 25 A standard ITU-T test sequence of video frames called Glasgow was used. The sequence had a QCIF resolution, that is a luminance picture size of 176 x 144 pixels and a chrominance picture size of 88 x 72 pixels.

- 30 The first 297 frames of the Glasgow sequence were compressed to produce a temporally scalable bitstream comprising INTRA (I) frames, INTER (P) frames and bi-directionally predicted (B) frames. The I and P frames are referred to as primary frames and the B frames are referred to as secondary frames. The frame rate of the original (default) sequence was 12.5 frames per second (fps) and so the compressed sequence corresponded to 23.68 seconds. Each frame was

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bitstream with another bitstream.

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The reordering algorithm used two values:

full-quality playback assuming a full rate channel; and

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for display as a picture (allowing, of course, for the time needed to receive the bits for a frame and to prepare them) and this time is compared with the time stamp of the frame. If the frame will be received after it is to be displayed, the buffering time is increased, if it is received in time then the next frame can be dealt with. Once  
 5 the algorithm has compared the likely times of receipt of all of the frames against their playback times and adjusted to buffering time accordingly so that they should all be received in time, the minimum buffering time has been calculated.

The algorithm was applied to a sequence having its original order. In this case,  
 10 FQFR is smaller than RQRR because in order to obtain full quality playback, you have to wait longer in starting playback if you have a reduced rate channel. The algorithm seeks to minimise the initial buffering time needed for full-quality playback with full channel throughput and at the same time for reduced-quality playback with a reduced channel throughput. In other words, the algorithm seeks  
 15 to find a value where FQFR and RQRR are approximately equal. It should be noted that the algorithm is not optimal, that is the algorithm does not necessarily re-order the sequence so that the minimum initial buffering delay is found. The algorithm consists of the following steps:

1. Calculate FQFR and RQRR.

20 2. If FQFR is greater than or equal to RQRR, stop re-ordering. The minimum initial buffering delay is the greater of FQFR or RQRR. Otherwise, continue with the next step.

3. Shift all primary frames towards the beginning of the sequence by one frame step. For example, if the initial sequence is:

25 I0, P1, P2, B3, P4, B5, P6, B7, P8, B9, P10, ...,

shifting changes the sequence to:

I0, P1, P2, P4, B3, P6, B5, P8, B7, P10, B9, ...

where I0 is the first frame. It can be seen that where a secondary frame is earlier in sequence than an adjacent primary frame, the positions of the primary and  
 30 secondary frames are exchanged.

4. Continue from step 1.

The buffering time needed to ensure pause-less playback in the reduced bit rate channel is greater for the original sequence than for the re-ordered sequence. It is desirable to have a short initial buffering time. The re-ordering process makes a trade-off between the minimum buffering time needed for a full-rate channel and the minimum buffering time needed for expected reduced-rate channels. Without re-ordering, less initial buffering time would have been needed to provide full-quality playback in the full-rate channel. However, without re-ordering, more initial buffering time would have been required to provide reduced-quality playback in the reduced-rate channel.

In the example, safety times were also investigated. Initial buffering time was set to be the minimum for reduced quality playback of the re-ordered sequence at 25% secondary bit rate. Figure 12 shows how safety time varies with playback of the 25-ordered sequence and the original order sequence. It shows that throughout playback, the 25-ordered sequence (represented by the upper line) has a longer safety time than the original order sequence.

Results were also obtained for the 50% and 75% secondary bit rates. The average safety times (in seconds) for all primary frames in a sequence are shown in the following table for all three reduced bit rates. The safety times are characterised by minimum safety time and average safety time.

		25%	50%	75%
Minimum	Re-ordered	4.79	2.51	1.06
	Default	3.46	1.63	0.64
	Difference	1.33	0.88	0.42
Average	Re-ordered	5.90	3.51	2.12
	Default	4.66	2.83	1.84
	Difference	1.24	0.67	0.28

It can be seen that both the minimum and average safety times of the re-ordered sequences are longer than those of the original sequence.

In contrast, set out below is a table showing the required buffering delays in seconds for full-quality playback at different average bit rates. This is to minimise pauses in playback.

	25%	50%	75%
Re-ordered	10.04	6.03	3.53
Default	7.02	4.64	3.02

It can be seen that the re-ordered sequence needs a longer buffering delay for full-quality playback because the secondary frames are transmitted out of their natural order and time is required to put all the frames back into their natural order. However, this is not important because the invention sets out to provide reduced quality playback with minimal initial buffering delay. To illustrate this, the following table shows the buffering delays required to ensure pause-less reduced quality playback:

	25%	50%	75%
Re-ordered	5.28	3.49	2.47
Default	6.91	4.46	2.84

These results show that where the bit rate of the transmission channel is reduced, the invented method requires a shorter buffering delay before playback can be started.

The effect of the invention is illustrated in Figure 13 which shows playback of the 25-ordered sequence at 25% bit rate (that is  $A+0.25B$  as discussed above). Playback times, that is the times at which the frames are intended or supposed to be played to a viewer, are represented as a straight line. To be played in sequence, without requiring pauses, each frame must be received before its playback time. Any symbol located below the playback time line represents a

frame which is received before its scheduled playback time and any symbol located above the playback time line represents a frame which is received after its scheduled playback time. In the example of Figure 13, the primary frames were all received in time to be played and the secondary frames were all received too late.

5 Therefore the secondary frames were all discarded without being played.

It should be understood that if the sequence had not been re-ordered, there is a likelihood that some of the primary frames would have arrived after their playback times and so would have required playback to be paused (since a primary frame cannot be discarded without also discarding all of the frames predicted from it).

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Figure 14 shows how the 25-ordered sequence is received and played when the channel throughput is not reduced. It can be seen that all frames are received in time to be played. Again, the upper line represents when secondary frames are received and the lower line represents when primary frames are received.

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Of course, the exact appearance of Figures 13 and 14 depends on the nature of the original sequence, as well as the bit rate and the re-ordering that is performed. Differences can be expected in other systems which re-order data in different ways.

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In use, the invention can be applied on-line to prepare a bitstream for transmission at some future point. For example, assuming that there may be considerable variation in the available bandwidth of a given transmission channel, the algorithm may be applied to a bitstream and re-ordered sequences may be calculated for several alternative bandwidths, such as 25%, 50% and 75%. In transmission, if problems are encountered, for example with pausing, the re-ordered sequence currently being used may be changed to one better suited to a lower available bandwidth. Equally, if transmission is proceeding better than expected, a sequence better suited for a higher bandwidth may be used. The invention may check the progress of transmission and/or playback a number of times during transmission and adjust the sequence accordingly. It is advantageous if this

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adjustment can be programmed dynamically whilst transmission is occurring. In this way, intermittent problems can be dealt with.

5 The part of the invention which re-orders a bitstream is located in an editor or in a server. The part of the invention which recovers the order is located in a client.

10 A further improvement is provided if the proportion of bits belonging to the primary frames is reduced by replacing some of the primary frames with B-frames. This reduces the compression efficiency slightly because B-frames are predicted from reference frames (P-frames) which are further away in time and so the B-frames and P-frames are less alike. This yields a poorer predicted B-frame and so more bits must be used to code the corresponding prediction error image. Since in this arrangement there are now more bits belonging to secondary frames which can be discarded, re-ordering the frames can provide even better protection against  
15 transmission delays and pauses.

20 One way of looking at the invention is to say that it sacrifices the safety times of the secondary frames in order to maintain maximum safety times for the primary frames.

The invention is particularly suitable for streaming-type multimedia retrieval.

25 While preferred embodiments of the invention have been shown and described, it will be understood that such embodiments are described by way of example only. For example, although the invention has been described applied to a temporally scalable bitstream, it can also be applied to other types of scalability, including spatial, spectral and SNR. Numerous variations, changes and substitutions will occur to those skilled in the art without departing from the scope of the present invention. Accordingly, it is intended that the following claims cover all such  
30 variations as fall within the spirit and scope of the invention.

## CLAIMS

1. A server for transmitting a data signal having a sequence of data units in a predetermined order over a transmission link the data units being sent in an order determined by their relative importance rather than their predetermined order.
2. A server according to claim 1 in which the data units represent a base layer and at least one enhancement layer.
3. A server according to claim 1 comprising re-ordering means to change the order of the data units.
4. A server according to claim 1 in which the data signal is scalable.
5. A server according to claim 4 in which the signal is scalable in a domain selected from a group consisting of the temporal, the spatial, the spectral and the SNR domains.
6. A server according to claim 1 which comprises an editor for providing the data signal.
7. A server according to claim 1 in which the data signal represents a sequence of pictures to produce a moving image.
8. A server according to claim 7 in which the data signal represents a video sequence.
9. A server according to claim 1 in which the data signal comprises multimedia data.
10. A data transmission system for transmitting a data signal having a sequence of data units in a predetermined order over a transmission link between a data



source and a data sink the data units being sent in an order determined by their relative importance rather than their predetermined order.

11. A transmission system according to claim 10 in which the data units each  
5 comprise a base layer and at least one enhancement layer and when re-ordered the base layer of a particular data unit has a greater safety time than of the or each enhancement layer of the particular data unit.
12. A transmission system according to claim 10 in which the source is a server.
- 10 13. A transmission system according to claim 10 in which the source is an editor.
14. A transmission system according to claim 10 in which the sink is a client.
- 15 15. A transmission system according to claim 10 in which the sink is a mobile terminal.
16. A transmission system according to claim 10 in which the sink is a mobile  
20 telephone.
17. A transmission system according to claim 10 in which means are provided to check the progress of transmission and to change the order being used to one better suited to available bandwidth.
- 25 ~~18. A method of transmitting a data signal having a sequence of data units in a predetermined order over a transmission link between a data source and a data sink comprising the step of sending the data units in an order determined by their relative importance rather than their predetermined order.~~
- 30 19. A method according to claim 18 in which the data units are returned to their original sequence once they have been transmitted over the transmission link.

20. A method according to claim 18 in which the progress of transmission is checked and the order being used is changed to one better suited to available bandwidth.

5 21. A computer program product stored on a computer usable medium comprising computer readable program means for causing transmission of a data signal having a sequence of data units in a predetermined order over a transmission link the data units being sent in an order determined by their relative importance rather than their predetermined order.

10 22. A computer program product according to claim 21 comprising a server.

23. A computer program product according to claim 21 comprising an editor for providing a scalable data signal.

15 24. A computer program product according to claim 22 comprising re-ordering means for providing the layers of the or each data units with different safety times.

20 25. A data signal having a sequence of data units for transmission over a transmission link between a data source and a data sink the data units being in an order determined by their relative importance rather than their predetermined order.

25 26. A method of controlling transmission of a data signal having a sequence of data units in a predetermined order over a transmission link between a data source and a data sink, the method comprising the steps of:  
monitoring the transmission of the data units in order to determine the condition of the transmission link;  
30 providing a control signal to the data source to change the order of the data units in response to the condition of the transmission link not being adequate;

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transmitting the data units from the data source to the data sink; and

- 1

A data transmission system (10) is provided for transmitting a data signal between a server (12, 14, 16, 18) and a communications terminal (20, 22) over a transmission link having a variable bandwidth. The data signal is a scalable compressed multimedia clip comprising a sequence of images. Each image has a base layer and a number of enhancement layers. To cope with variability in the bandwidth, the sequence is re-ordered so that the base layers have greater safety times than the enhancement layers. This allows all of the base layers to arrive in time to be played at the expense of the enhancement layers.

(Figure 9)

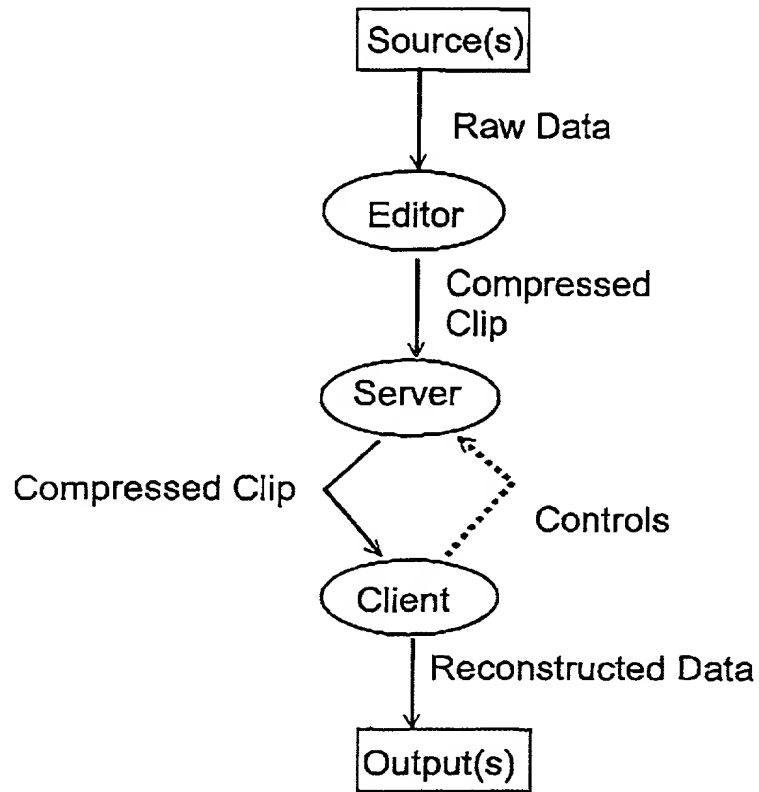


Fig. 1

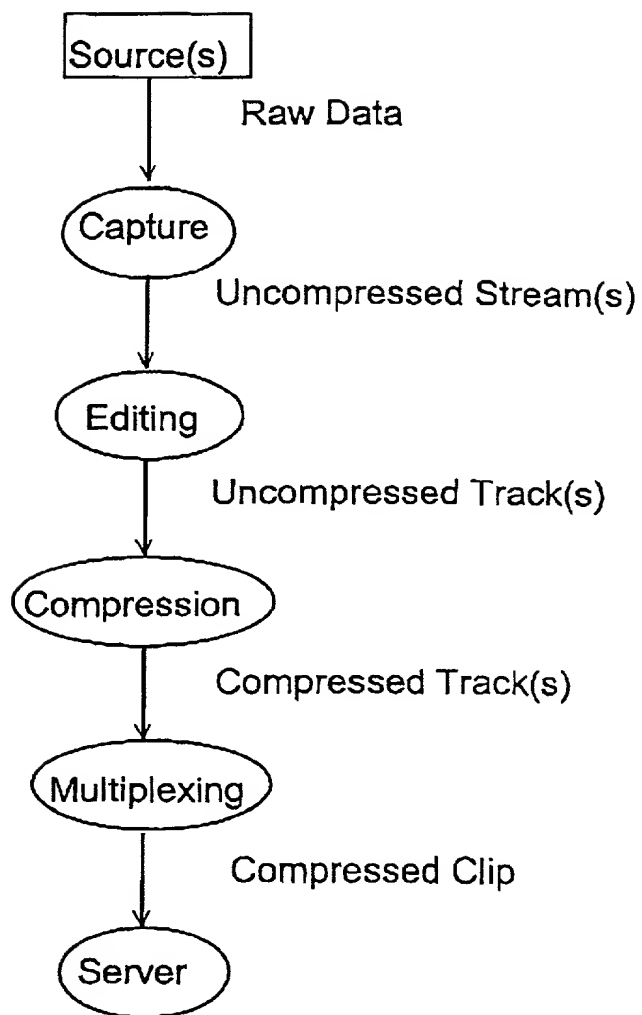


Fig. 2

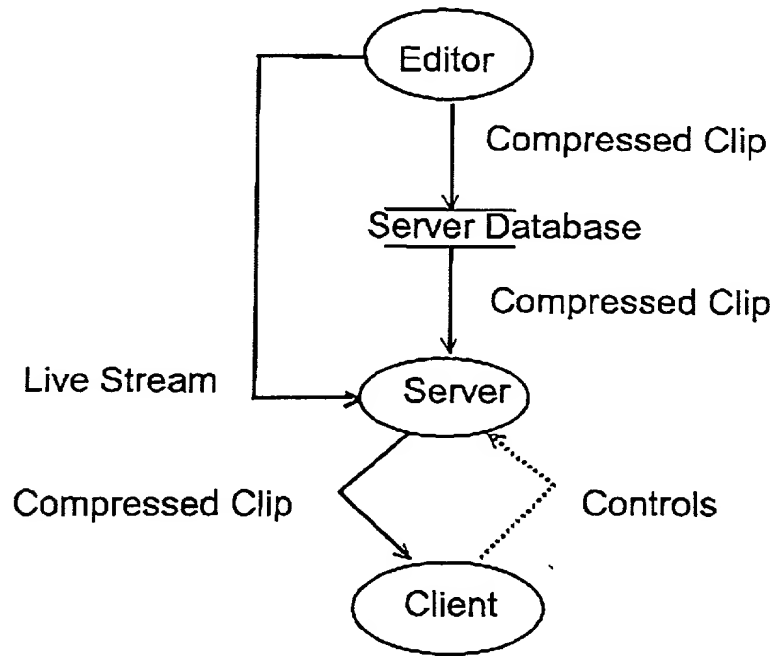


Fig. 3

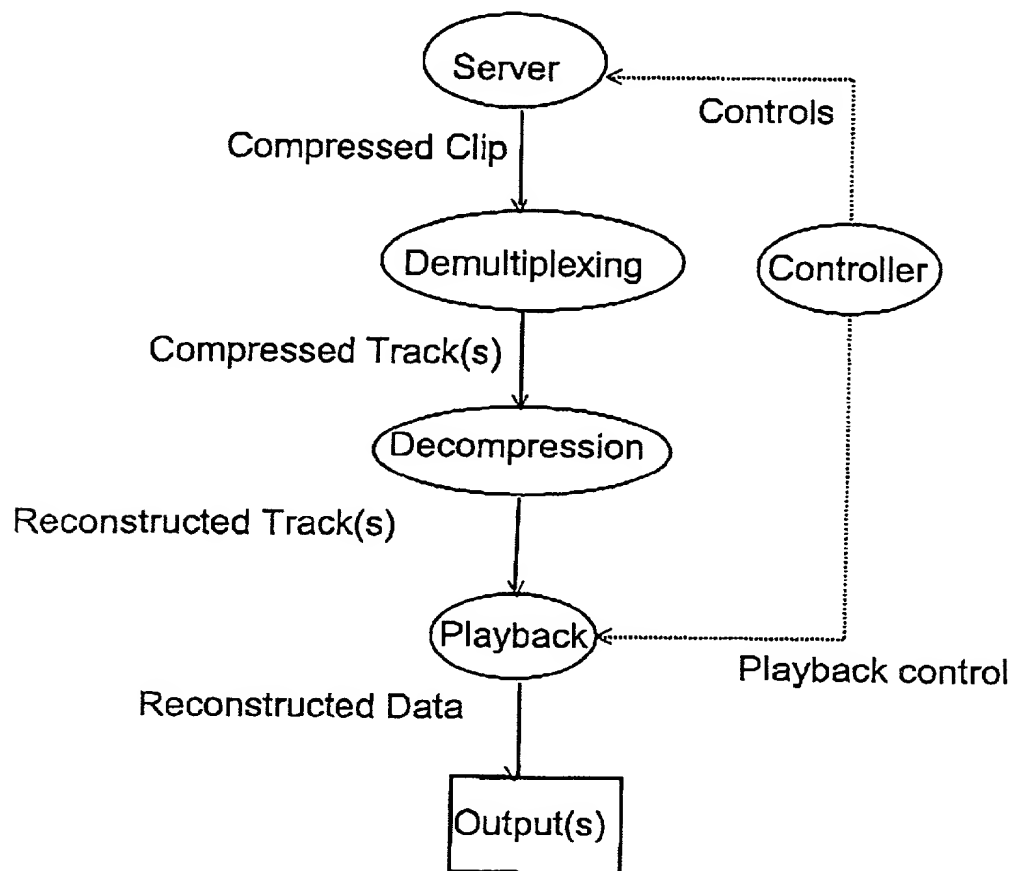


Fig. 4



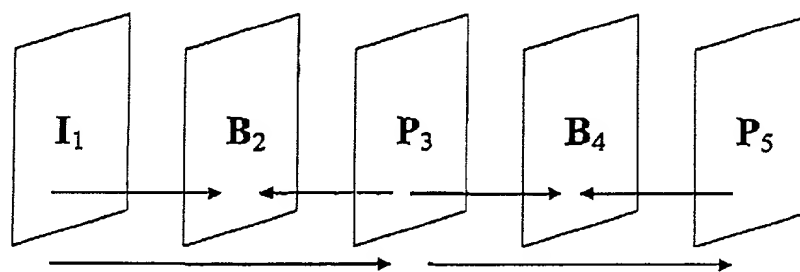


Fig. 5

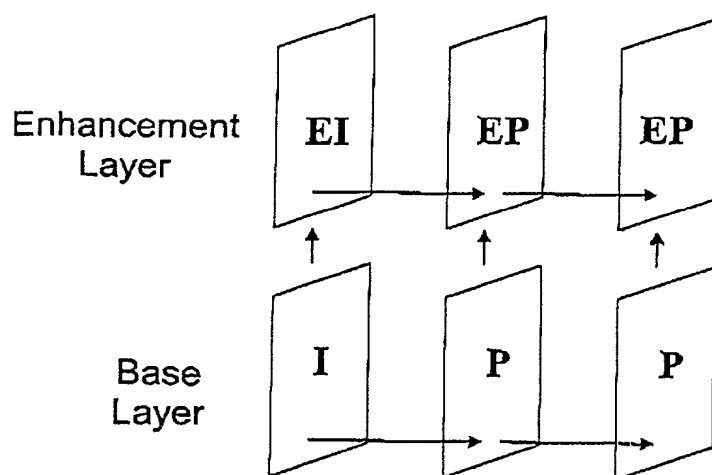


Fig. 6

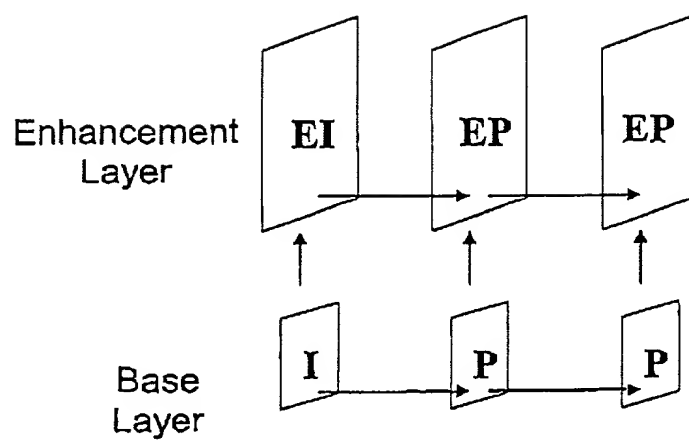


Fig. 7

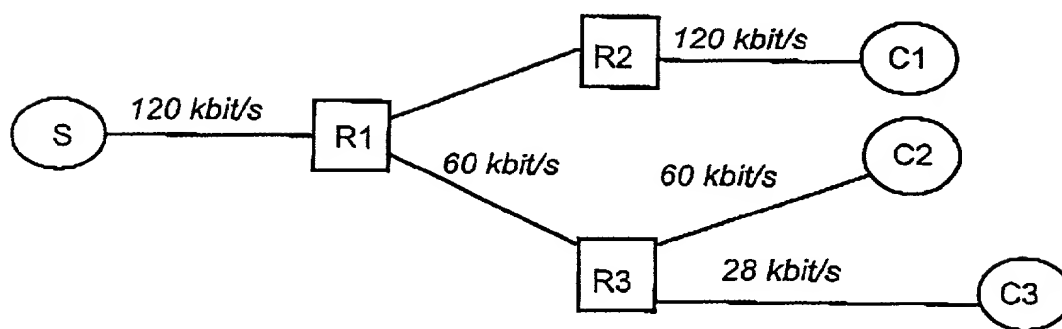


Fig. 8

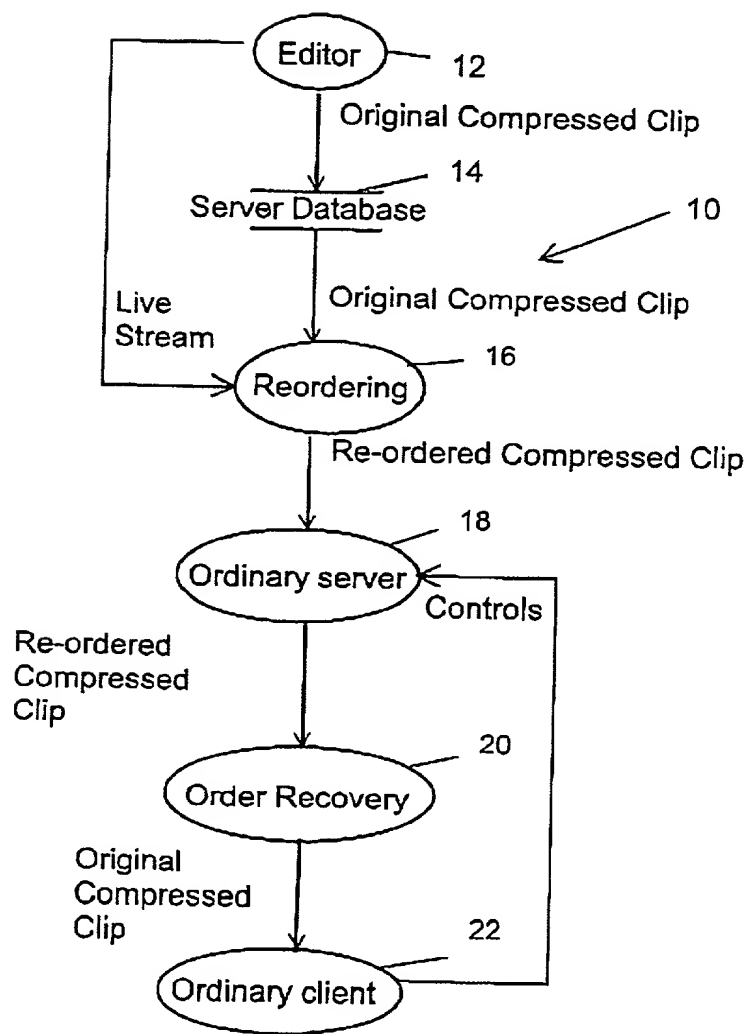


Fig. 9

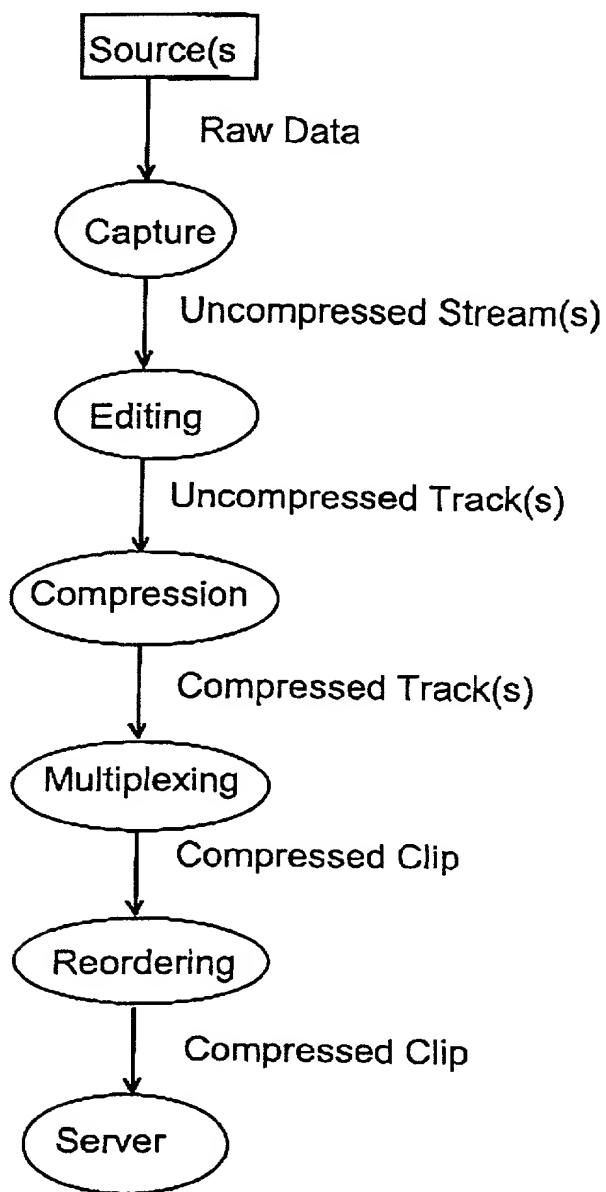


Fig. 10

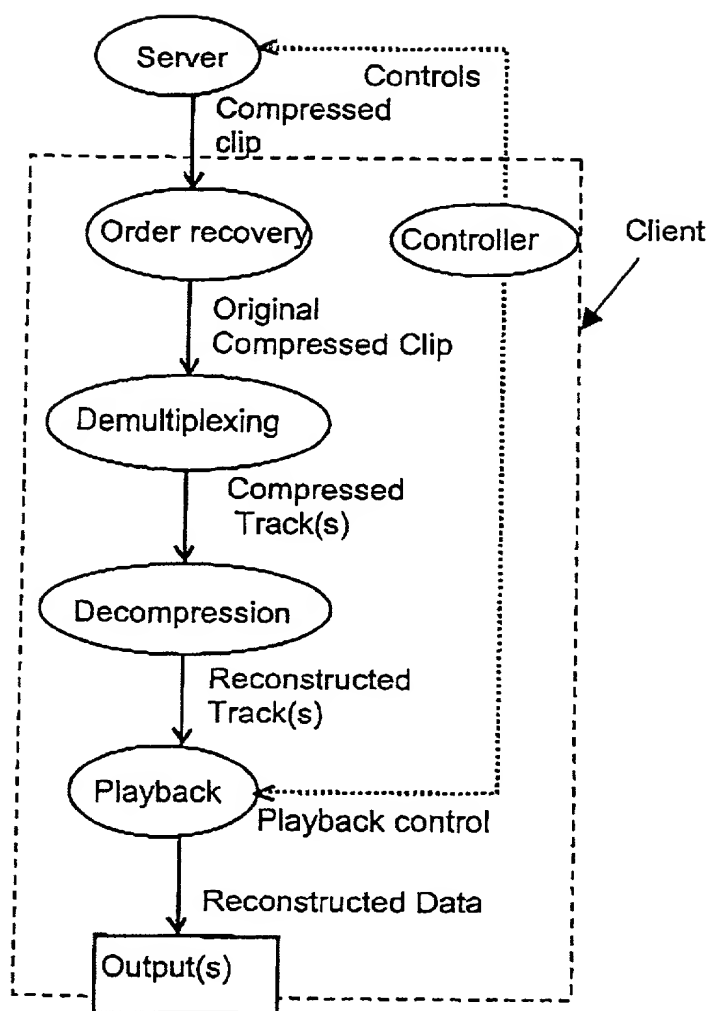


Fig. 11

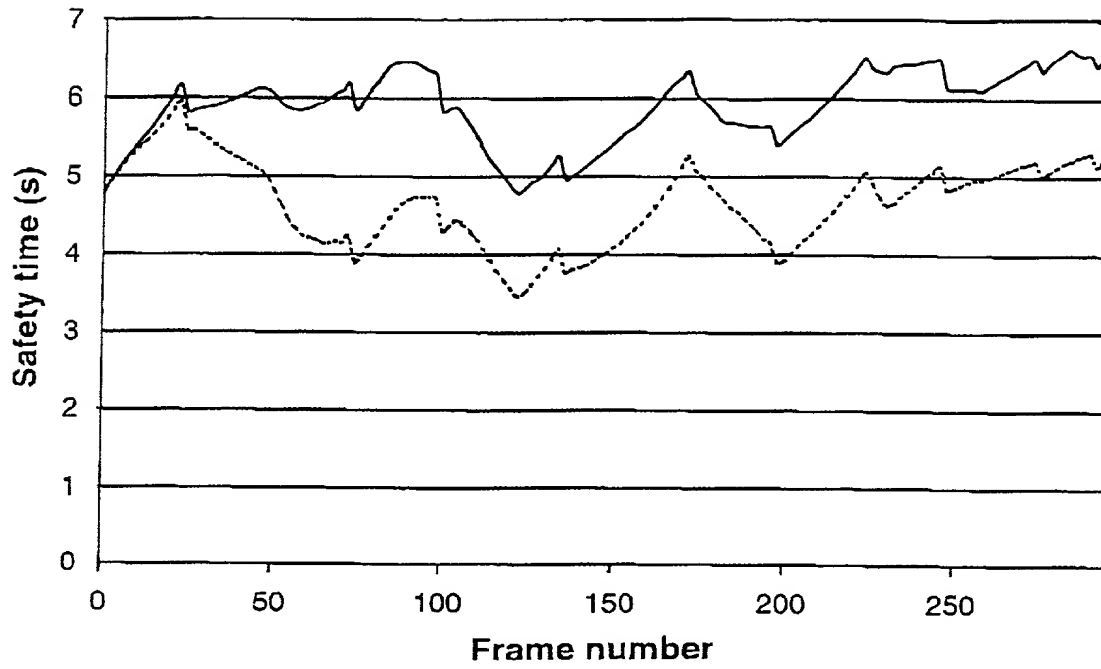


Fig. 12

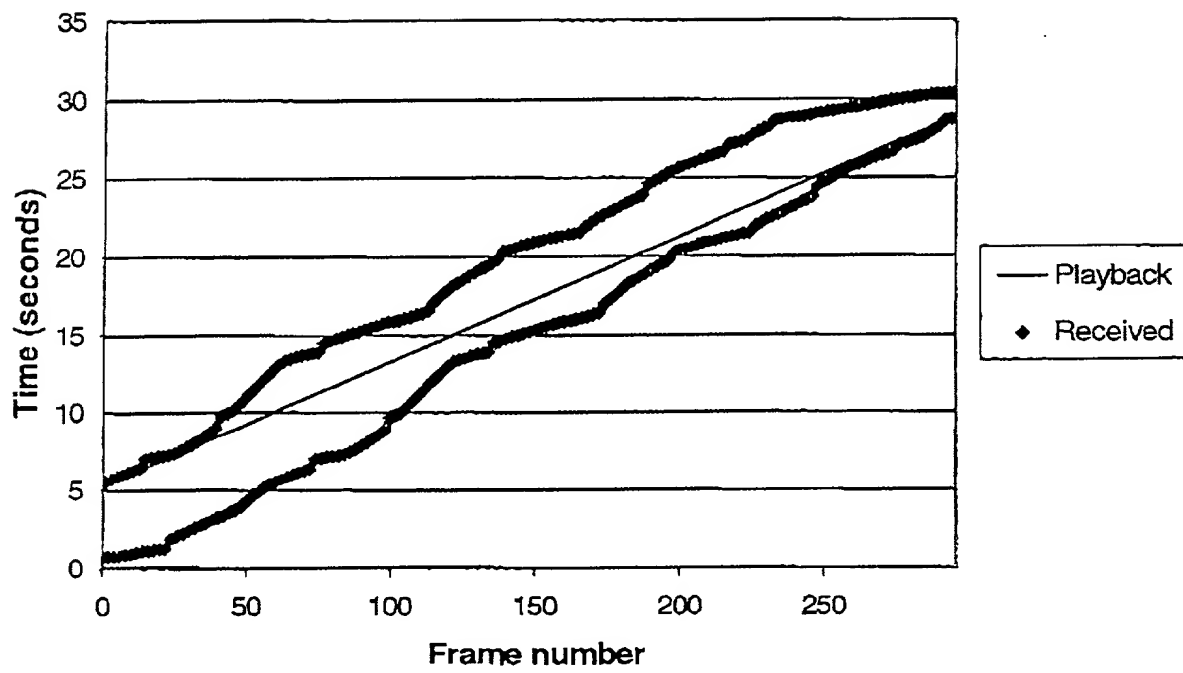


Fig. 13

12/13

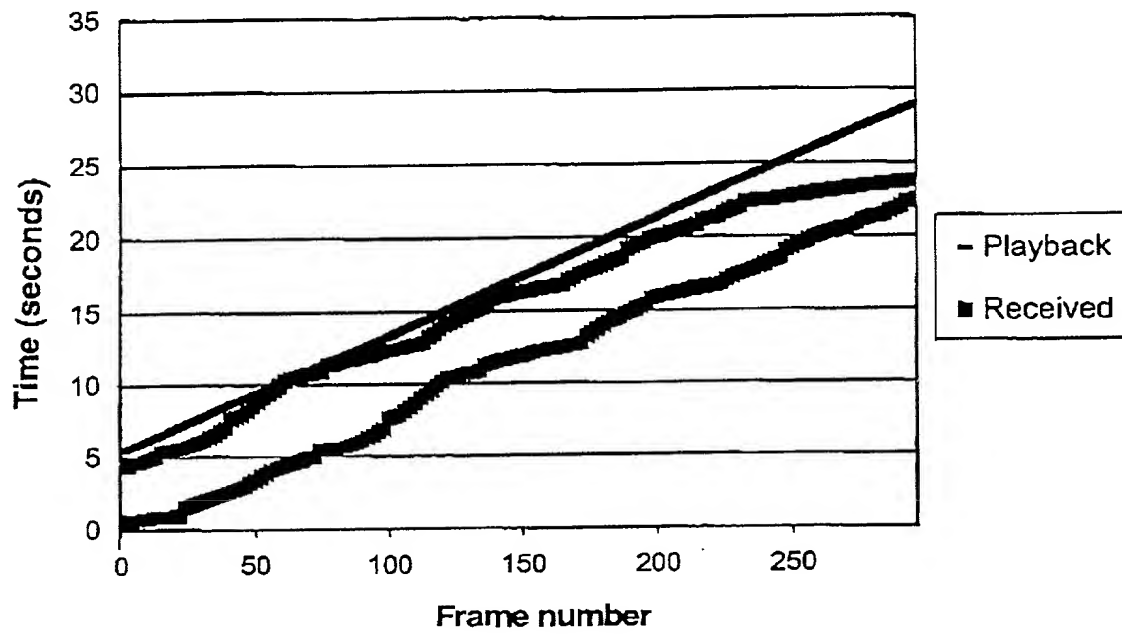


Fig. 14



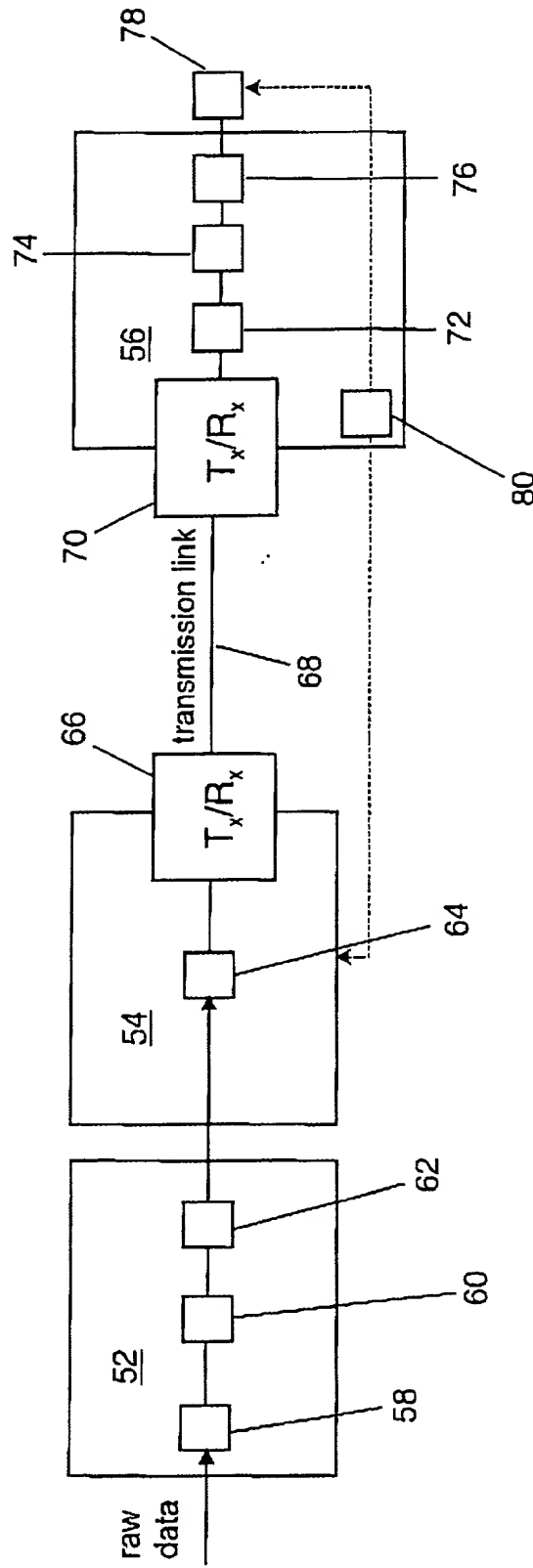


Fig. 15